(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 0 814 456 A2

(12) # EUROPEAN PATENT APPLICATION

(43) Date of publication:
29.12.1997 Bulletin 1997/52

(51) Int Cl.[6]: G10K 11/178

(21) Application number: 97302679.2

(22) Date of filing: 18.04.1997

(84) Designated Contracting States:
DE FR GB

(30) Priority: 17.06.1996 US 673458

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(54) **Active noise or vibration control (ANVC) system and method including enhanced reference signals**

(57) An ANVC system (**50**) and method of enhancing the reference (input) signal (**51**) provided from a reference sensor (**52**) in a feedforward-type Active Noise or Vibration Control (ANVC) system (**50**). Preferably, an Adaptive Line Enhancer (ALE) (**54**) is provided in the input path for reducing the uncorrelated noise present in the reference signal (**51**). In one aspect, a tone(s) present can be enhanced by reducing broadband uncorrelated noise. In another aspect, the broadband input to the ANVC control (**58**) can be enhanced by eliminating uncorrelated tone(s). The filter structure used in the ALE (**54**) may include IIR or FIR and the algorithm used to update the ALE filters may include LMS, RLS, or GMV. Parametric and adaptive inverse ALEs (**69**) and (**80**) are also described. In alternate embodiments, multiple ALEs are arranged in cascaded or parallel relationship. Further, the tonal output (**22**) of the ALE (**54**) may be used as an input to auxiliary components such as Engine Vibration Monitors (EVMs). The ALEs are beneficial in both tonal and broadband ANVC systems (**50**).

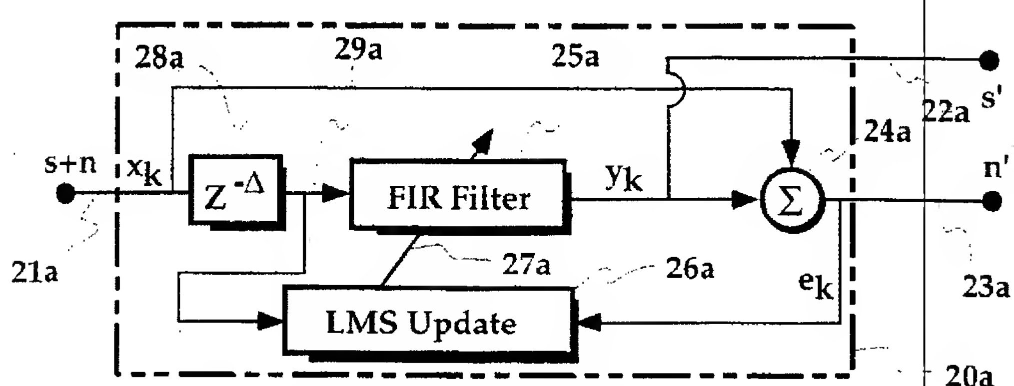


Fig. 1 Prior Art

EP 0 814 456 A2

Printed by Jouve, 75001 PARIS (FR)

Description

Field of the Invention

The invention relates to the area of active noise and vibration control. Specifically, the invention relates to a feed-forward active noise and vibration control system using a reference input.

Background of the Invention

Active noise and vibration control (ANVC) systems generally utilize input or reference sensors which provide signals indicative of the source of disturbance (noise or vibration), error sensors which provide signals indicative of residual noise or vibration to be canceled, and adaptive processing of the input and error signals in order to arrive at output signals. These output signals drive output transducers thereby producing antinoise or antivibration with the result of reducing the residuals at the error sensors, thereby reducing the noise or vibration thereat. It is known that the Single-Input Single-Output (hereinafter SISO) ANVC variant is a subset of the more general Multi-Input Multi-Output (hereinafter MIMO) ANVC system. See IEEE paper, Vol. ASSP-35, No. 10, Oct. 1987, by Elliott, Stothers, and Nelson, entitled "A Multiple Error LMS Algorithm and Its Application to the Active Control of Sound and Vibration" for a description of one MIMO system. It is desired to have a reference signal which is only indicative of the source of disturbance to be canceled, if possible. However, in many real-world applications, the input signal may be corrupted with background noise which is not correlated with the disturbance to be canceled. A corrupted input signal is likely to corrupt the output signals, as any noise contained therein may be passed on directly to the output transducers (speakers, Active Vibration Absorbers (AVAs), or other actuators), and may even be amplified. The presence of this background noise detracts from the effectiveness of the ANVC system and contributes to increase the overall level of noise detected by the error sensors.

Therefore, there is a recognized need for a method of reducing the background noise present in input signals, i. e., a method of increasing the signal-to-noise ratio. In particular, there is a dire need for such a method for application to tonal ANVC systems where a reference signal contains one or more dominant tones that are being controlled, yet the input signal including the tones, also includes unwanted broadband noise.

Adaptive line enhancement (ALE) methods are known for separating a signal into a periodic component signal and a random component signal. One known example application of ALE techniques is for canceling the maternal heartbeat in fetal electrocardiography as described in *Adaptive Signal Processing*, pp. 334-337, by Widrow and Stearns, 1985. Another known example application of ALE techniques is for removing periodic interference from training noise in an ANVC system identification process as described in *Investigations in Active Line Enhancer Techniques* by Shawn Steenhagen, MS Thesis, Univ. of Wisconsin, Madison, 1993.

Prior Art Fig. 1 describes one conventional prior art ALE implementation. The conventional Finite Impulse Response (FIR) ALE **20a** processes an ALE input signal **s+n, 21a,** containing the periodic component signal **s** plus the random component signal **n** with the objective of separating the periodic component signal from the random component signal. The ALE input signal **s+n** is delayed Δ>0 samples by the delay operator **28a** to produce the delayed input signal **29a**. The delayed input signal is processed by an FIR filter **25a** to produce an estimate **s'** for the periodic component signal **22a**. The periodic component signal estimate is subtracted from the input signal by the error summation block **24a** to produce an estimate **n'** for the random component signal **23a**. One or both of these component estimates may be used depending on the requirements of the application. The random component signal **23a** is also interpreted as an ALE error signal **ek**, given by

$$e_k = x_k - y_k$$

where **xk** is the ALE input **21a**, and **yk** is the adaptive FIR filter output **22a** given by

$$y_k = \sum_{i=0}^{N_A-1} A[i]_k x_{k-i-\Delta}$$

The ALE input signal **21a** and the broadband component estimate signal **n'** are processed by an adaptive update block **26a**, using a gradient descent method such as LMS

$$A[i]_{k+1} = A[i]_k + \mu_A e_k x_{k-i-\Delta} \qquad i = 0,(N_A-1)$$

to provide updated filter coefficients **27a** to the FIR filter **25a**.

Prior Art Fig. 2 describes a second conventional prior art ALE implementation. The conventional Infinite Impulse Response (IIR) ALE **30b** processes an ALE input signal **s+n, 21b,** containing the periodic component signal **s** plus the random component signal **n** with the objective of separating the periodic component signal from the random component signal. The ALE input signal **s+n** is delayed Δ>0 samples by the delay operator **35b** to produce the delayed input signal **36b**. The delayed input signal is processed by an FIR A-filter **Ak, 31b,** to produce an output **39b.** The estimate **s'** for the periodic component signal **22b** is processed by an FIR B-filter **Bk, 32b,** to produce an output **40b**. The A-filter output is combined with the B-filter output in the path summation block **41b** to generate the periodic component signal estimate **22b**. The periodic signal component estimate is given by

$$y_k = \sum_{i=0}^{N_A-1} A[i]_k x_{k-i-\Delta} - \sum_{j=1}^{N_B-1} B[j]_k y_{k-j}$$

The periodic component signal estimate is subtracted from the ALE input signal by the error summation block **24b** to produce an estimate **n'** for the broadband component signal **23b**. One or both of these component estimates may be used depending on the requirements of the application. The ALE input signal **21b** and the broadband component estimate signal **n'** are convolved by an A-filter correlator block **33b** which is used in the LMS gradient descent method to provide A-filter adjustments **34b** to the FIR A-filter **31b**. The periodic component signal estimate **s'** and the broadband component estimate signal **n'** are convolved by a B-filter correlator block **37b** which is used in the LMS gradient descent method to provide B-filter adjustments **38b** to the FIR B-filter **32b**. These filter coefficient updates are given by

$$A[i]_{k+1} = A[i]_k + \mu_A e_k x_{k-i-\Delta} \qquad i = 0,(N_A - 1)$$

$$B[j]_{k+1} = B[j]_k + \mu_B e_k y_{k-j} \qquad j = 1, (N_B - 1)$$

when the LMS algorithm is used. However, none of the above address the need for an ANVC system with an enhanced reference signal.

Summary of the Invention

Therefore, in light of the problems associated with prior art ANVC systems identified above, the present invention, in one aspect thereof, is directed to a method and means for reducing unwanted background noise that is present in a reference (input) signal in a feedforward-type ANVC system. In particular, the present invention is directed to adaptive gradient descent means for reducing the noise present in the reference (input) signal of an ANVC system. More specifically, an Adaptive Line Enhancer (ALE) is preferably placed in the input path of the ANVC system. The adaptive means, such as an ALE, for enhancing the reference signal can reduce broadband noise from the input signal if there are one or more dominant tones therein to be controlled, or conversely, tones may be reduced if a broadband input is required for the input to the ANVC system.

The ALE may include IIR, FIR, parametric or adaptive inverse implementations, or the like. The ALEs may be adaptively controlled via any of the known gradient descent algorithms. The use of ALEs in the input path is beneficial to any of the known ANVC systems, including Active Noise Control (ANC), Active Structural Control (ASC), and Active Isolation Control (AIC). Further, the ALE enhanced reference signal described herein can be used with any of the known ANV control processes, such as those including FIR and IIR adaptive filters. In another aspect, multiple ALEs may be used in parallel or cascaded relationship to provide, for example, a reference signal with contributions from multiple tones. Further, the enhanced reference signal provided by the ALE, besides being used as an input to the control process, may be used as an input to auxiliary components, such as an Engine Vibration Monitor (EVM) which is used for monitoring, and/or, displaying the vibration of aircraft engines.

The main advantage of the invention is that it provides a reference (input) signal to the ANVC system that has substantially higher Signal-To-Noise (SNR) ratio.

It is another advantage of the invention that the power requirements to drive the output transducers can be reduced.

It is another advantage of the invention that it may reduce the computational requirements for the adaptive control of the ANVC system, for example, the number of taps in a multi-tap control algorithm may be reduced.

It is another advantage of the invention that it may enhance the performance and convergence of the ANVC system.

The abovementioned and further novel aspects, features and advantages of the invention will be apparent from the accompanying descriptions of the preferred embodiments and attached drawings.

Brief Description of the Drawings

The accompanying drawings which form a part of the specification, illustrate several key embodiments of the present invention. The drawings and description together, serve to fully explain the invention. In the drawings,

Fig. 1 is a schematic diagram of a prior art FIR ALE,

Fig. 2 is a schematic diagram of a prior art IIR ALE,

Fig. 3 is a block schematic view of an Active Noise Control (ANC) system using a microphone input sensor, an ALE for reducing noise in the reference signal which is uncorrelated with the disturbance to be canceled and a loudspeaker as an output transducer,

Fig. 4 is a schematic view of another Active Noise Control (ANC) system using an accelerometer input sensor, an ALE in the input path, and a loudspeaker as the output transducer,

Fig. 5 is a schematic view of an Active Structural Control (ASC) system using an accelerometer input sensor, an ALE on the input, and an Active Vibration Absorber (AVA) as the output transducer,

Fig. 6 is a schematic view of an Active Isolation Control (AIC) system using an accelerometer input sensor, an ALE in the input path, and an active mounting as the output transducer,

Fig. 7 is a schematic block diagram of one embodiment of the present invention ANVC system including a FIR ALE in the input path and an FIR process control for tonal control,

Fig. 8 is a schematic block diagram of another embodiment of the present invention ANVC system including a IIR ALE in the input path and an FIR process control for tonal control,

Fig. 9 is a schematic block diagram of another embodiment of the present invention ANVC system including a IIR ALE in the input path and an IIR process control for tonal control,

Fig. 10 is a schematic block diagram of a parametric ALE,

Fig. 11 is a schematic block diagram of an adaptive inverse ALE,

Fig. 12 is a schematic block diagram of another embodiment of the present invention ANVC system including a parametric ALE in the input path and an FIR process control for tonal control,

Fig. 13 is a schematic block diagram of another embodiment of the present invention ANVC system including a adaptive inverse ALE in the input path and an FIR process control for tonal control,

Fig. 14 is a schematic block diagram of another embodiment of the present invention ANVC system including a FIR ALE and FIR process control for broadband control,

Fig. 15 is a schematic block diagram of another embodiment of the present invention ANVC system including multiple Low-Order IIR ALEs,

Fig. 16 is a schematic block diagram of another embodiment of the present invention ANVC system including a High-Order FIR ALE,

Fig. 17 is a schematic block diagram of another embodiment of the present invention ANVC system including multiple cascaded ALEs,

Fig. 18 is a schematic block diagram of another embodiment of the present invention ANVC system including using the ALEs output for an input to an auxiliary component,

Fig. 19 is a figure illustrating a reference signal having multiple tones and broadband background noise, and

Fig. 20 is a figure illustrating an output signal from the Band Pass Filter (BPF) and also from the ALE within the input path showing the resulting further reduction in unwanted broadband background noise in the reference signal.

Detailed Description of the Preferred Embodiments

With reference to the figures herein, where like reference characters are employed where possible to indicate like parts, there is shown in **Fig. 3**, an ANVC system, and in particular, an ANC system **50c** comprising an input sensor **52c** for providing a reference signal indicative of the disturbance acoustic noise or vibration causing the disturbing acoustic noise. In the ANC case, the disturbance may be an acoustic noise emanating from a noise source, such as an aircraft engine or the like. In most applications, the reference signal generally will include some unwanted noise therein. By the term noise, what is referred to is background noise which is uncorrelated with the disturbance which is sought to be canceled. In the ANC system **50c**, adaptive means are provided for reducing the noise present in the reference signal.

In particular, an Adaptive Line Enhancer (ALE) **54c**, which includes an adaptive filter and update means for updating the coefficients or weights of the adaptive filter, is preferably is used. An error sensor **62c** is provided for generating an error signal indicative of the residual acoustic noise at the point adjacent where a quiet zone is desired. The means for processing the error signal and the reference signal and producing an output signal is provided in the control process **58c,** in this case, ANC control, which includes a filter taking the form of a IIR or FIR filter structure with adaptive feedforward control. An output transducer **60c**, in this case, a loudspeaker, is dynamically driven responsive to the output signal. The output transducer **60c** produces antinoise which preferably minimizes the noise at the point of interest to produce a quiet zone. The controller **56c** includes both the adaptive means, such as ALE **54c**, for reducing the unwanted noise on the input signal and the control process **58c** therewithin. If should be understood that the reference enhancement means is based upon an adaptive gradient descent method and is generally implemented within the software. It should also be understood that there may be an optional filtering/conditioning step before the reference signal is provided to the ALE **54c**.

Fig. 4 represents another ANC system **50d** which is identical fo the system of **Fig. 3** except that the reference sensor is an accelerometer **52d**. One ANC system for which fhe ALE used within the inpuf path as described herein may be useful is discussed in commonly assigned US Application Serial No. 08/553,227 to G. Billoud entitled "Active Noise Control System for Closed Spaces Such As Aircraft Cabins" filed Sept. 25, 1995. Other ANC systems are described in US Pat. No. 4,562,589 to Warnaka et al. entitled "Active Attenuation of Noise in a Closed Structure" and US Pat. No. 4,473,906 to Warnaka et al. entitled "Active Acoustic Attenuator."

Fig. 5 represents another ANVC system, and in particular, an ASC system **50e** which is identical to the system of **Fig. 4** except that the output transducer is an AVA **60e**. Active systems including AVAs are described in PCT Patent Application Serial No. PCT/US95/13610 entitled "Active Systems and Devices Including Active Vibration Absorbers (AVAs)" and US Pat. No. 4,715,559 to Fuller entitled "Apparatus and Method For Global Noise Reduction." In aircraft ASC systems, AVAs are attached directly to the interior surface of the aircraft's fuselage and dynamically shake the fuselage wall to generate canceling noise in the aircraft's cabin.

Fig. 6 represents another ANVC system, and in particular, an AIC system **50f** which is identical to the system of **Fig. 5** except that the output transducer is an active mount **60f**. Active mounts are taught in commonly assigned US Pat. No. 5,174,552 to Hodgson et al. entitled "Fluid Mount with Active Control" and US Pat. App. Serial No. 08/260,945 entitled "Active Mounts for Aircraft Engines." Further descriptions may be found in a Lord paper entitled "Frequency-Shaped Control of Active Isolators" by D. A. Hodgson. Active mounts **60f** attach between an engine and the structure the engine is attached to and are dynamically actuated (driven) to control vibration therebetween or noise at a remote location.

Fig. 7 represents a detailed block diagram of a tonal ANVC system **50g** including a reference sensor **52g**, an optional band pass filter **53g**, an ALE **54g**, and output transducer **60g**, and error sensor **62g** and an adaptive control process **58g** which may include system identification, hereinafter referred to as ID **42g**. The ID **42g** may be accomplished in an on-line or off-line fashion. Further, filtering or other signal conditioning is commonly used on the output signal and error signal paths, however, they are not shown for the sake of clarity in all figures described herein. The ALE **54g** herein includes a FIR filter structure as is described fully with reference to **Fig. 1**. The reference signal **51g** from reference sensor **52g** is preferably band-pass filtered and received at the ALE input **21g**. The output of the ALE **22g** is provided to the control process **58g** for the ANVC system **50g**. In this embodiment, the ALE output **22g** is comprised of the periodic component of the reference signal **51g**, i.e., the one or more tones that are indicative of the

noise or vibration **49g** generated by the source of disturbance **48g**.

In this embodiment, the control process **58g** is achieved by an FIR filter **59g** including update means, such as filtered-x LMS, or the like. A further discussion of a Single-Input Single-Output (SISO) Filtered-x LMS control with system ID can be found in "Adaptive Control of a Two-Stage Vibration Isolator Mount" by S. D. Sommerfeldt and J. Tichy, Journal of the Acoustical Society of America, Vol. 88, No. 2, pp. 938-944, 1990. A further discussion of Multiple-Input Multiple-Output (MIMO) systems including FIR filters is described in US Pat. No. 5,170,433 to Elliott et al. entitled "Active Vibration Control." In General, the ALE output **22g** is filtered through an error path model **63g** (sometimes referred to as the X filter) representative of the transfer function between each output transducer **60g** and error sensor **62g** pair. The filtered ALE output is referred to as the filtered regressor **65g**. Error information in line **64g** represents the error signal **55g** and any filtered training noise from training block **42g**. Error information **64g** and the filtered regressor **65g** are inputted to an update method and means **61g**, such as Filtered-x LMS, for determining the new filter weights to be passed to the FIR control filter **59g**. The output **22g** from the ALE **54g** is filtered by the control filter **59g** to arrive at the output signal **57g** used to drive the output transducer **60g**, e.g. a loudspeaker, AVA, active mount or the like. It should be understood that the use of the ALE **54g** in the input path provides a cleaner reference signal to the control filter **59g**, therefore, the output signal **57g** to the output transducer **60g** is also cleaner and the ANVC system **50g** will do a better job at cancellation of the noise or vibration.

Fig. 8 represents another tonal feedforward ANVC system **50h** including a reference sensor **52h**, an adaptive gradient descent means for reducing the uncorrelated noise present in the reference signal **51h**, such as an ALE **54h**, an adaptive control process **58h** including an FIR filter **59h** which may include system identification **42h**, an output transducer **60h**, and an error sensor **62h**. The ALE **54h** used herein is fully described with reference to **Fig. 2** and represents a IIR filter ALE. Again, the output **22h** from the IIR ALE **54h** is used as an input to the control process **58h** which is identical to that described with reference to **Fig. 7**.

Fig. 9 represents another tonal feedforward ANVC system **50j** including identical elements as described with reference to **Fig. 8** except that the control process **58j** includes a IIR filter instead of a FIR filter. Therefore, the ANVC system **50j** is a combination of a IIR control filter process **58j** and a IIR ALE **54j**. Control process **58j** may include system ID **42j** which may be implemented in an on-line or off-line fashion. IIR control filter structures are described in US Pat. No. 4,677,676 to Eriksson entitled "Active Attenuation System with On-Line Modeling of Speaker, Error Path, and Feedback Path" and US Pat. No. 4,677,677 to Eriksson entitled "Active Sound Attenuation System with On-Line Adaptive Feedback Cancellation."

Fig. 10 is a schematic showing one embodiment of a parametric IIR ALE. In this configuration, the adaptive filter is a IIR filter similar to the prior art **Fig. 2**; however, the parametric IIR ALE described herein does not require a delay operation as in **35b**, and the filter coefficient update process is greatly simplified with a constrained adaptation process due to the parametization. The parametric IIR ALE **69m** processes an ALE input signal **s+n, 21m,** containing the periodic component signal **s** plus the random component signal **n** with the objective of separating the periodic component signal from the random component signal. The ALE input signal is processed by an FIR A-filter **A(ωk), 31m**, to produce an output **39m.** The estimate s' for the periodic component signal **22m** is processed by a FIR B-filter **B(ωk), 32m**, to produce an output **40m**. The A-filter output is combined with the B-filter output in the path summation block **41m** to generate the periodic component signal estimate **22m.** The periodic component signal estimate is subtracted from the ALE input signal by the error summation block **24m** to produce an estimate **n'** for the broadband component signal **23m.** One or both of these component estimates may be used depending on the requirements of the application.

Each of the IIR filter coefficients in **A(ωk)** and **B(ωh)** are explicitly parametized by the center frequency ωk. This parametization is accomplished by first selecting a desired frequency response for the IIR filter. A preferred response is a second-order band-pass filter which is given by

$$\left[\frac{(BW)s}{s^2+(BW)s+\omega_o^2}\right] \underset{\text{Bilinear Transform}}{\overset{\text{Pre-Warped}}{\Leftrightarrow}} \left[\frac{A_0\left(1-z^{-2}\right)}{1+B_1z^{-1}+B_2z^{-2}}\right]$$

The filter bandwidth (*BW*) and the sharpness of resonance (*Q*) are given by

$$BW = 2\xi\omega_o \quad \Leftrightarrow \quad Q=\left(\frac{\omega_o}{BW}\right)=\left(\frac{1}{2\xi}\right)$$

where ξ is the damping ratio, and ω_o is the center frequency. Using the pre-warped bilinear transform, the three non-

zero digital filter coefficients may be analytically expressed in terms of the center frequency and bandwidth. This parametization is given by

$$A_o(\omega_k)=\left(\frac{BW\sin(\omega_k T)}{2\omega_k+BW\sin(\omega_k T)}\right)$$

$$B_1(\omega_k)=\left(\frac{-4\omega_k\cos(\omega_k T)}{2\omega_k+BW\sin(\omega_k T)}\right)$$

$$B_2(\omega_k)=\left(\frac{2\omega_k-BW\sin(\omega_k T)}{2\omega_k+BW\sin(\omega_k T)}\right)$$

where *T* is the sample period, and ωk is the time varying center frequency. For a constant bandwidth, the digital filter coefficients are parametized by center frequency only.

There are many methods available for adaptively updating the center frequency. One preferred approach is to use an LMS gradient descent method which minimizes the instantaneous squared ALE error **ek**, represented by **23m**, which is equivalent to **n'**. The ALE error is given by

$$e_k = x_k - y_k$$

where **xk** is the ALE input **21m**, and **yk** is the adaptive IIR filter output **22m**. For the second-order band-pass filter response, the adaptive IIR filter output is given by

$$y_k = A_0(\omega_k)x_k - A_0(\omega_k)x_{k-2} - B_1(\omega_k)y_{k-1} - B_2(\omega_k)y_{k-2}$$

Taking the derivative of the instantaneous squared ALE error with respect to center frequency ωk

$$J=e_k^2 \quad\Rightarrow\quad -\nabla_k=\left(\frac{\partial J}{\partial\omega_k}\right)=\left(\frac{\partial J}{\partial A(\omega_k)}\right)\left(\frac{\partial A(\omega_k)}{\partial\omega_k}\right)+\left(\frac{\partial J}{\partial B(\omega_k)}\right)\left(\frac{\partial B(\omega_k)}{\partial\omega_k}\right)$$

the update expression for the center frequency becomes

$$\omega_{k+1}=\omega_k+\mu e_k\left[\left(\frac{\partial A_o(\omega_k)}{\partial\omega_k}\right)x_k-\left(\frac{\partial A_o(\omega_k)}{\partial\omega_k}\right)x_{k-2}-\left(\frac{\partial B_1(\omega_k)}{\partial\omega_k}\right)y_{k-1}-\left(\frac{\partial B_2(\omega_k)}{\partial\omega_k}\right)y_{k-2}\right]$$

where μ is a constant step size which controls the rate of convergence. The derivatives are also analytically available for the second-order band-pass filter response, and are given by

$$\left(\frac{\partial A_0(\omega_k)}{\partial \omega_k}\right) = 2BW\left(\frac{\omega_k T\cos(\omega_k T) - \sin(\omega_k T)}{(2\omega_k + BW\sin(\omega_k T))^2}\right)$$

$$\left(\frac{\partial B_1(\omega_k)}{\partial \omega_k}\right) = 2\left(\frac{2BW\omega_k T + 4\omega_k^2 T\sin(\omega_k T) - BW\sin(2\omega_k T)}{(2\omega_k + BW\sin(\omega_k T))^2}\right)$$

$$\left(\frac{\partial B_2(\omega_k)}{\partial \omega_k}\right) = -2\left(\frac{\partial A_0(\omega_k)}{\partial \omega_k}\right)$$

The parametric FIR filter coefficients **31m** and **32m** are updated with each new estimate of the center frequency **77m**. The center frequency is updated by the center frequency update **76m** wherein the gradient estimate **75m** is added to the previous center frequency value. The gradient estimate is the output of a gradient summation block **74m** which sums the A-filter gradient contribution **71m** and the B-filter gradient contribution **73m**. The ALE input signal is convolved with **n'** in the A-filter correlator block **33m** which produces an A-filter convolution vector **78m**. The A-filter convolution vector is multiplied by the gradient of the A-filter coefficients with respect to center frequency, in the A-filter parametric gradient product block **70m**, which produces the output A-filter gradient contribution **71m**. The periodic component signal estimate **s'** is convolved with **n'** in the B-filter correlator block **37m** which produces a B-filter convolution vector **79m**. The B-filter convolution vector is multiplied by the gradient of the B-filter coefficients with respect to center frequency, in the B-filter parametric gradient product block **72m**, which produces the output B-filter gradient contribution **73m**.

The parametric IIR ALE may be further simplified in several ways. One improvement to the invention is to restrict the operational range of the center frequency, which is easily accomplished when the center frequency is adapted explicitly. The restriction is accomplished by not allowing the adaptation process to select a center frequency outside of some prescribed range. Upon inspection of the partial derivatives of the filter coefficients with respect to center frequency, it may be observed that some of the derivatives are several orders of magnitude smaller than the others. These small derivatives may be assumed zero and the corresponding parameter values may be assumed constant over the restricted frequency range. The transcendental parametric expressions for the coefficients whose derivatives are not negligible may also be preferably replaced with polynomial curve fit, or other expressions which are simpler to evaluate in real time.

Fig. 11 is a schematic showing one embodiment for an adaptive inverse ALE. In this configuration, the adaptive filter is an FIR filter similar to the prior art **Fig. 1**; however, the adaptive inverse ALE uses the adaptive filter coefficients in a novel way for separating the periodic component signal from the random component signal. The adaptive inverse ALE, **80p**, processes an ALE input signal **21p** where reference numerals **21p, 28p, 29p, 34p, 33p, 25p**, and **24p** are exactly as described in reference to **Fig. 1**. Signal **81p** is an auxiliary periodic component signal and signal **82p** is an auxiliary random component signal which are only used in the adaptation of the FIR A-filter **Ak**.

The ALE input signal is further processed by a IIR filter which is constructed from the modified FIR filter coefficients **Ak**, and an additional filter Gk as described below. The ALE error signal ek is given by

$$e_k = x_k - y_k$$

where **xk** is the ALE input **21p**, and **yk** is the adaptive FIR filter output **81p**. The preferred FIR A-filter is a two-coefficient filter whose output is given by

$$y_k = A[1]_k x_{k-\Delta} + A[2]_k x_{k-1-\Delta}$$

The LMS gradient descent update for each of these two coefficients is given by

$$A[1]_{k+1} = A[1]_k + \mu_A e_k x_{k-\Delta}$$

$$A[2]_{k+1} = A[2]_k + \mu_A e_k x_{k-1-\Delta}$$

In order to understand how the adaptive inverse ALE operates, we first look at an example input signal which is a pure tone as represented in complex notation by

$$x_k = X\cos(\omega_o kT + \phi) = \mathrm{RE}\left[Xe^{j(\omega_o kT+\phi)}\right]$$

where X is an arbitrary non-zero amplitude, ω_o is the radian frequency, k is the sample index, T is the sample period, and ϕ is an arbitrary phase angle. Substituting this result into the expression for the ALE error signal defined above

$$e_k = \mathrm{RE}\left[X\left(e^{j(\omega_o kT+\phi)} - A[1]e^{j(\omega_o(k-\Delta)T+\phi)} - A[2]e^{j(\omega_o(k-1-\Delta)T+\phi)}\right)\right]$$

$$e_k = \mathrm{RE}\left[\left(Xe^{j(\omega_o kT+\phi)}\right)\left(1 - A[1]e^{-j\omega_o\Delta T} - A[2]e^{-j\omega_o(1+\Delta)T}\right)\right]$$

$$e_k = \mathrm{RE}\left[\left(Xe^{j(\omega_o kT+\phi)}\right)\left(1 - e^{-j\omega_o\Delta T}\left(A[1] + A[2]e^{-j\omega_o T}\right)\right)\right]$$

Under perfect convergence conditions, the ALE error signal will be zero and thus

$$e_k = 0 \quad \Rightarrow \quad \left(A[1] + A[2]e^{-j\omega_o T}\right) = e^{j\omega_o\Delta T}$$

This equation may be solved for the ideal or optimal A-filter coefficients as

$$A[1] = \cos(\omega_o\Delta T) + \left(\frac{\sin(\omega_o\Delta T)}{\sin(\omega_o T)}\right)\cos(\omega_o T)$$

$$A[2] = -\left(\frac{\sin(\omega_o\Delta T)}{\sin(\omega_o T)}\right)$$

It can be shown that the LMS gradient descent update described above will drive the A-filter coefficients toward the optimal values given above. A further simplification can be achieved by selecting Δ=1. The ALE input signal is input to a FIR feedforward filter **Gk, 31p**, which produces a G-filter output **83p**. The periodic component signal estimate **22p** is produced from the path summation **41p** whose inputs are the G-filter output and the modified A-filter output **84p**. In the preferred embodiment, the G-filter is a single coefficient which appropriately scales the ALE input signal. The signal **s'** is delayed by the delay block **28p'** to produce the delayed output **29p'** which is input to the modified A-filter **85p**. The modified A-filter produces the modified A-filter output **84p**. If the A-filter coefficients from **25p** were used directly in **85p** without modification, the resulting digital IIR filter would have undamped poles at the frequency of the input excitation. In practice, damping is added to the poles. In the preferred embodiment, the transfer function *G(z)* of the resulting IIR filter is given by

$$\Delta = 1 \quad \Rightarrow \quad G(z) = \left[\frac{G_k}{1 - \gamma A[1]z^{-1} - \gamma^2 A[2]z^{-2}}\right]$$

and the filter output **rk**, which is equivalent to **s'**, is given by

$$r_k = G_k x_k + \gamma A[1]_k r_{k-\Delta} + \gamma^2 A[2]_k r_{k-1-\Delta}$$

where 0<<γ<1. The poles of the resulting IIR filter are

$$z = \gamma[\cos(\omega_o T) \pm j\sin(\omega_o T)]$$

$$\|z\| = \gamma = e^{-\xi\omega_0 T} = e^{-(^{BW}/_2)T}$$

and thus we see that the modification of scaling **A[1]** by γ and scaling **A[2]** by γ^2 preserves the center frequency while adding damping. The G-filter must generally be selected such that the resulting IIR filter has the appropriate gain at the center frequency. In the preferred embodiment, the gain is unity and the resulting scalar G-filter coefficient must be

$$\|G(j\omega_o)\| = 1 \Rightarrow G_k = (1 - \gamma)\sqrt{1 - 2\gamma\cos(2\omega_o T) + \gamma^2}$$

thus we see that the G-filter is parametized by center frequency.

The estimate **s'** for the periodic component signal **22p** is subtracted from the ALE input **21p** in the error summation **24p'** to produce an estimate for the random component signal **23p**. One or both of these component estimates may be used depending on the requirements of the application.

The adaptive inverse ALE may be further simplified in several ways. By choosing Δ=1, A[2]=-1 it does not require adaptation, and $A[1]=2\cos(\omega o T)$. Since A**[1]** is parametrically related to the center frequency ωo, the adaptive update expression may be written in terms of an adaptive center frequency from which A**[1]** is then computed using the known parametric relationship. The scalar G-filter coefficient may then easily be computed using its known parametric relationship. One improvement to this aspect of the invention is to restrict the operational range of the center frequency which is easily accomplished when the center frequency is adapted explicitly. The restriction is accomplished by not allowing the adaptation process to select a center frequency outside of some prescribed range. When the operating frequency range is restricted, the transcendental parametric expressions for A[1] and the G-filter coefficient may be preferably replaced with polynomial curve fit or other expressions which are simpler to evaluate in real time.

Fig. 12 represents another tonal feedforward ANVC system **50m** including a reference sensor **52m**, adaptive gradient descent means for reducing the uncorrelated noise present in the reference signal **51m**, such as ALE **54m**, an adaptive control process **58m** including a FIR filter **59m** which may include system identification **42m**, an output transducer **60m**, and an error sensor **62m**. The ALE 54m used herein receives its ALE input **21m** in the form of the band-pass-filtered reference signal. The ALE output **22m** which represents a more refined tonal signal, as compared to the reference signal **51m** and the band-pass-filtered reference signal, is provided to the control process **58m**. The ALE **54m** is fully described with reference to **Fig. 10** and represents a parametric ALE. The simulated performance of this parametric embodiment is illustrated in **Fig. 20**. It should be understood that, although a FIR control process **58m** is shown, a IIR control process as described with reference to **Fig. 9** could also be employed in combination with the parametric ALE **54m**.

Fig. 13 represents another tonal feedforward ANVC system **50p** identical to that described with reference to **Fig. 12** except that the ALE **54p** is an adaptive inverse ALE. The ALE **54p** used herein is fully described with reference to **Fig. 11** and represents an adaptive inverse FIR ALE. It should be understood that although a FIR control process **58p** is shown, a IIR control process as described with reference to **Fig. 9** could also be employed in combination with the adaptive inverse ALE **54p**. As described with previous embodiments, the ALE includes its ALE input **21p** and ALE output **22p** within the reference path and is used to provide a refined tonal signal with the broadband uncorrelated noise reduced.

Fig. 14 represents a broadband feedforward ANVC system **50r** including an ALE **54r**. The ANVC system is identical to that described with reference to **Fig. 7** except that the ALE **54r** is used to reduce unwanted uncorrelated periodic noise (tones) in the reference signal path and leave only the broadband noise, which is desired to be controlled. Since

broadband noise is desired to be reduced by the broadband control process, then any uncorrelated tones present in the reference signal will detract from the effectiveness of the broadband control. The ALE **54r** used herein is fully described with reference to **Fig. 1** and represents a conventional FIR ALE. It should be understood that although a FIR control process **58r** and conventional FIR ALE **54r** are shown, that other combinations are possible. For example, the output from the conventional IIR ALE **30b** in **Fig. 2** at **23b** may be used as a broadband input to any ANV control such as the FIR ANV Control described with reference to **Fig. 8**. Similarly, the broadband output from the parametric ALE **69m** described in **Fig. 10** may be the broadband input to an FIR control similar to that described with reference to **Fig. 12**. Finally, the broadband output from the adaptive inverse ALE **80p** at **23p (Fig. 10)** may be used as an input to an ANV control, such as the control process described with reference to **Fig. 13**.

Fig. 15 represents a tonal feedforward ANVC system **50t** including multiple ALEs **54t** and **54t'** providing multiple signals to the ANV control. The ANVC system **50t** generally represents a MIMO system for controlling multiple tones emanating from the source **58t** which produce unwanted noise or vibration at a point of interest, for example within an aircraft cabin. In this embodiment, the reference sensor **52t** picks up the disturbance signal containing therein the multiple tones to be controlled. The signal is band-pass filtered to separate into two frequency ranges, **fr1** and **fr2**, the signals containing the tones of interest. For example, BPF **53t** passes the lower range of frequencies and BPF **53t'** passes only the upper range of frequencies.

The ALEs **54t** and **54t'** in this embodiment are IIR ALEs as described with reference to **Fig. 8** and are arranged in a parallel relationship. The ALEs **54t** and **54t'** may also be low-order IIR, FIR, parametric, or adaptive inverse ALEs. Low-order implies they each have only a low number of taps. ALE **54t** passes a lower frequency tone, for example, a signal indicative of the **N1** engine rotation frequency of an aircraft engine to the ANV control **58t**. Similarly, ALE **54t'** passes a higher frequency tone, for example, a signal indicative of the **N2** engine rotation frequency of an aircraft engine to the ANV control **58t**. The ANV control process **58t** then drives output transducer(s) **60t** responsive to the error signals from the error sensor(s) **62t** to preferably cancel the unwanted tones present at the point of interest that are caused by the source **48t**. It should be understood that in the tonal case, the ALEs **54t** and **54t'** reject the uncorrelated broadband noise within each frequency range of operation **fr1** and **fr2** and thereby provide an enhanced signal representative of the tones of interest to the ANV control **58t**.

Fig. 16 represents a tonal feedforward ANVC system **50v** including a high-order FIR ALE **54v** providing a signal to the ANV control **58v**. By the term high-order it is meant that the FIR filter therein has a high number of taps.

Fig. 17 represents a tonal feedforward ANVC system **50w** including a pair of cascaded ALEs **54w** and **54w'** providing reference signals to the ANV control **58w**. In this embodiment, the first ALE **54w** is used to enhance only the first tone of interest, while the second tone is enhanced by the second ALE **54w'**.

Fig. 18 represents a tonal feedforward ANVC system **50z** including an ALE **54z** for providing a clean signal to the ANV control **58z**. In this embodiment, the output from the ALE **54z** is also used as an input to an auxiliary component **90z**. For example, the output from the ALE **54z** may be used in an auxiliary component **90z** such as an Engine Vibration Monitor (EVM) including a peak detect **92z**, signal processing means **94z**, and display means **96z**.

Fig. 19 and **Fig. 20** represent the raw reference signal **51m** from the reference sensor **52m** and the filtered output from the BPF **53m**, i.e., the ALE input **21m**, and the tonal-enhanced ALE output **22m** from the ALE **54m** for the parametric embodiment of **Fig. 12**. The raw reference signal **51m** provided from the reference sensor **52m** is indicative of a reference signal (vibration) that is picked up by an accelerometer mounted on a jet engine. The engine has multiple vibrations, for example, at **N1** and **N2**, which produce tonal noise within the aircraft's cabin. Present in the reference signal **51m** is unwanted broadband background and some periodic noise **66m**. It is desired to have a separated reference signal that has contributions only at the tones that are to be controlled. The cleaner (less broadband) the reference signal **51m** the better, as any broadband noise present in the input to the FIR control filter **59m** may be passed onto the output transducers **60m**. The ALE input **21m** represents the reference signal once it has been filtered through the band pass filter **53m**.

In this case, the range of the band pass filter **53m** is centered around the nominal of **N1** and rejects broadband noise below and above its operating range. The ALE **54m** further enhances the input signal **21m** and produces an ALE output signal **22m** which is significantly more indicative of the tone of interest, in this case **N1**. It should be understood that similar results may be obtained for the conventional FIR ALE of **Fig. 1**, the conventional IIR ALE of **Fig. 2** and the adaptive inverse ALE of **Fig. 11**. Through the use of an ALE on the input path in a feedforward ANV control system, similar results can be achieved if one desires to remove the tones and leave the broadband noise as is described with reference to **Fig. 14**. This may be desirable for a system where the input is from a road wheel in a vehicle and broadband road noise is being controlled in the vehicle compartment or cabin.

In summary, the present invention is directed to a method and means for reducing unwanted background noise present in a reference (input) signal of a feedforward-type ANVC system. More specifically, an Adaptive Line Enhancer (ALE) is preferably placed in the input path of the ANVC system and enhances the reference signal by reducing broadband noise or tone(s) contained therein. The ALE may include IIR, FIR, parametric or adaptive inverse implementations, or the like. Further, the use of ALEs in the input path is beneficial to Active Noise Control (ANC), Active Structural

Control (ASC), and Active Isolation Control (AIC) systems. In another aspect, multiple ALEs may be used in parallel or cascaded relationship and, further, the ALE output may be used as an input to auxiliary components, such as Engine Vibration Monitors (EVMs). EVMs are described in SAE paper 871732 entitled "The V-22 Vibration, Structural Life, and Engine Diagnostic System, VSLED" by M. J. Augustin and J. D. Phillips.

While the preferred embodiments of the present invention have been described in detail, various modifications, additions, alternatives, changes, and adaptations to the aforementioned may be made without departing from the spirit and scope of the present invention defined in the appended claims. It is intended that all such modifications, additions, alternatives, changes, and adaptations be considered part of the present invention.

Claims

1. A method of enhancing a reference signal in an active noise or vibration control system, comprising the steps of:

   (a) receiving said reference signal from a reference sensor for providing a reference signal indicative of a disturbance, said reference signal including uncorrelated noise therein,

   (b) reducing the noise included within said reference signal by adaptive gradient descent means,

   (c) providing an error signal indicative of one selected from a group consisting of a residual noise and a residual vibration,

   (d) processing said error signal and said reference signal and producing an output signal, and

   (e) driving an output transducer based upon said output signal.

2. A method of claim 1 wherein adaptive gradient descent means for reducing said uncorrelated noise present in said reference signal is an Adaptive Line Enhancer (ALE) including an adaptive filter and update means.

3. A method of claim 2 wherein said ALE removes said uncorrelated noise which is at least one tone present in said reference signal.

4. A method of claim 2 wherein said ALE removes said uncorrelated noise which is broadband noise present in said reference signal.

5. A method of claim 2 wherein said adaptive filter within said ALE is a IIR filter or an FIR filter.

6. A method of claim 2 wherein coefficients of said adaptive filter within said ALE are modified according to said update means which includes an adaptive algorithm.

7. A method of claim 6 wherein said adaptive algorithm is an algorithm selected from a group consisting of a Least Means Square (LMS) algorithm, a Recursive Least Squares (RLS) algorithm, and a Generalized Minimum Variance (GMV) algorithm.

8. A method of claim 2 wherein said ALE is one selected from the group consisting of a conventional IIR ALE and a conventional FIR ALE.

9. A method of claim 2 wherein said ALE is an adaptive inverse ALE or a parametric ALE.

10. A method of claim 2 wherein said adaptive means for reducing said uncorrelated noise present in said reference signal is comprised of multiple Adaptive Line Enhancers (ALE).

11. An active noise or vibration control system of claim 10 wherein said multiple Adaptive Line Enhancers (ALE) are arranged in one selected from a group consisting of a cascaded relationship and a parallel relationship.

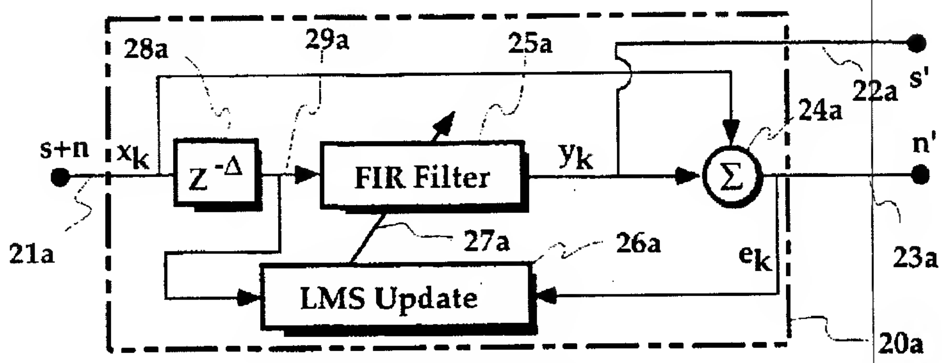


Fig. 1 Prior Art

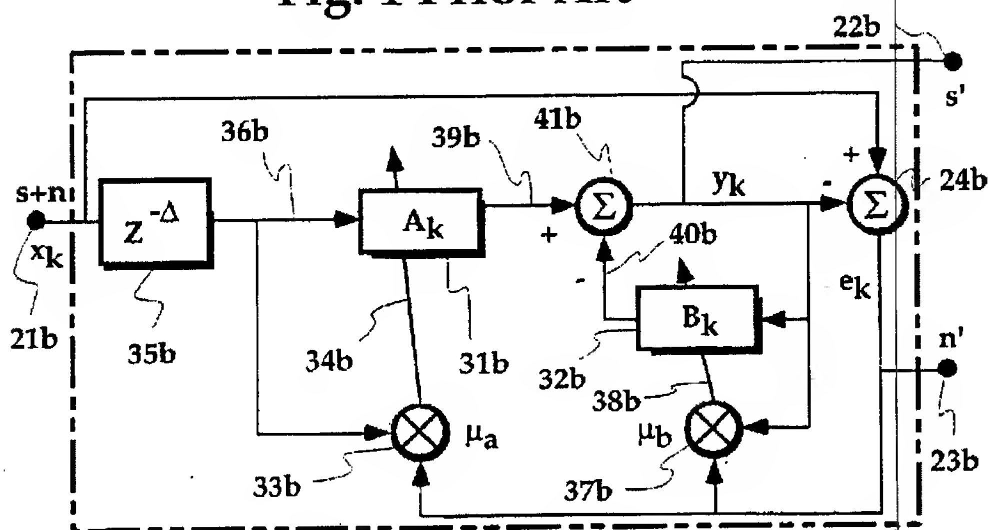


Fig. 2 Prior Art

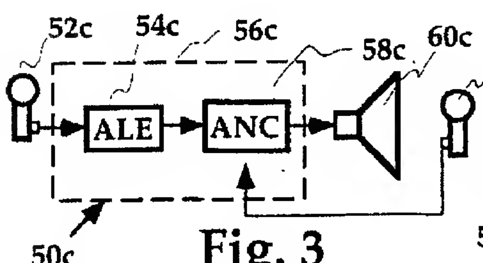


Fig. 3

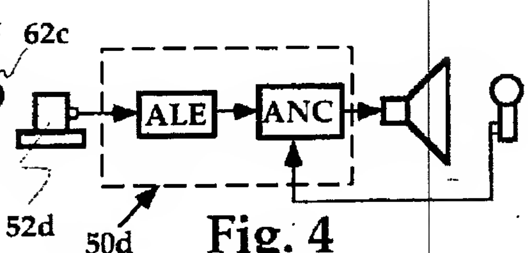


Fig. 4

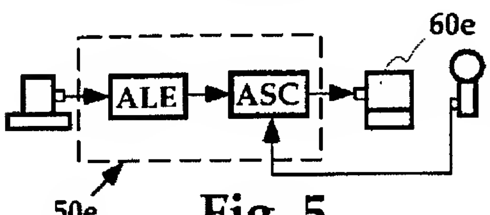


Fig. 5

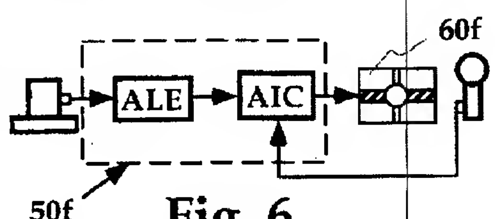
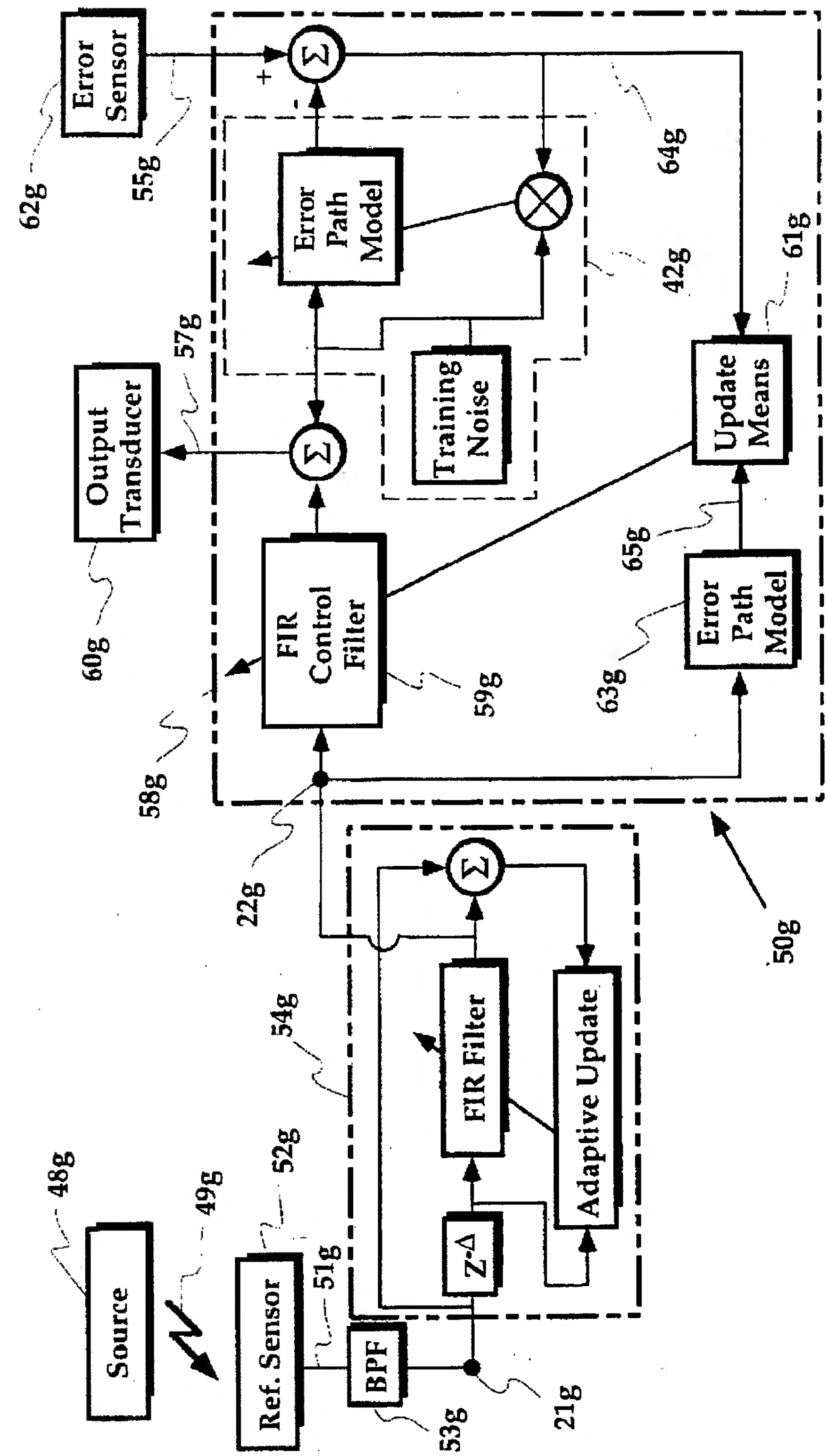


Fig. 6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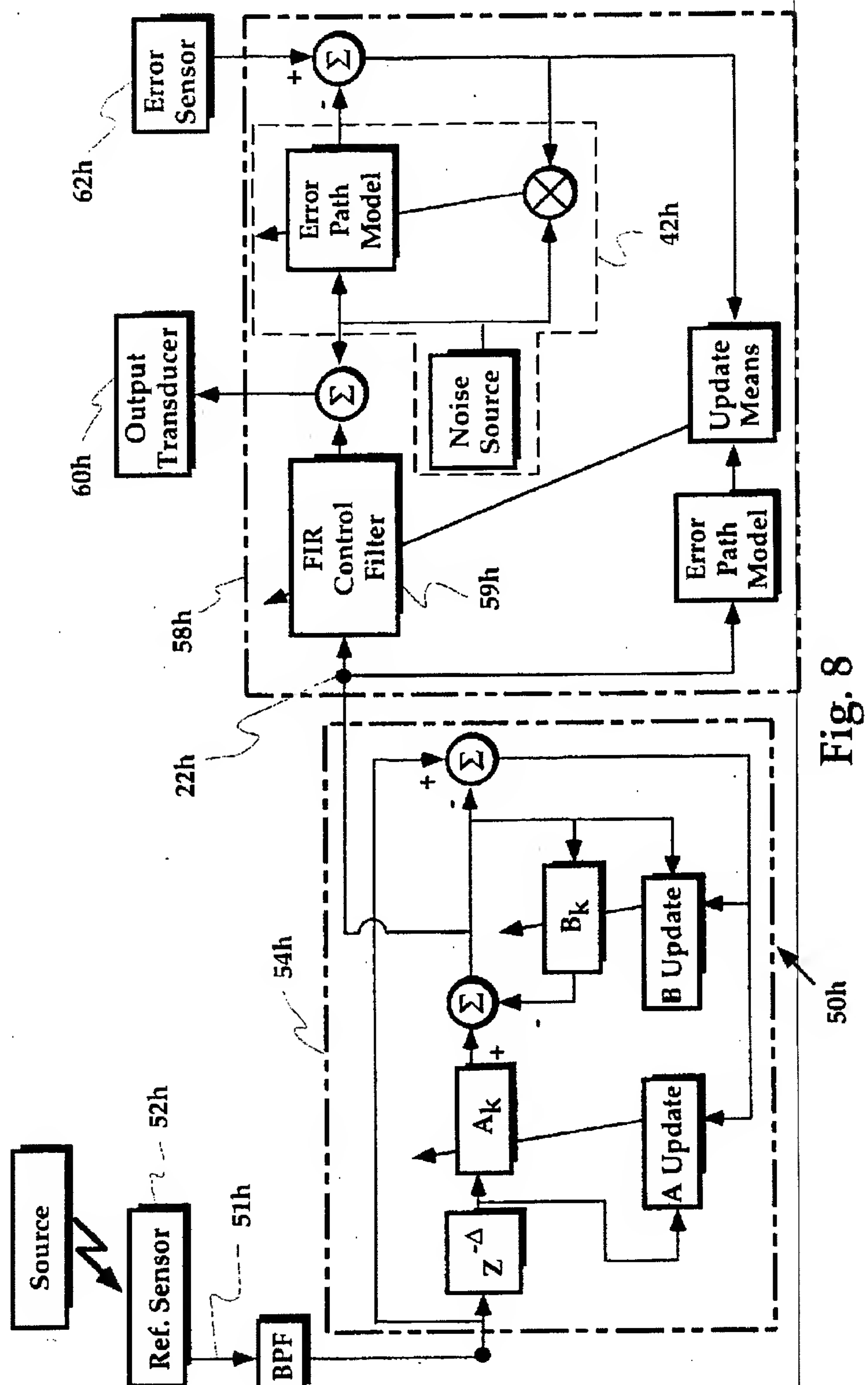
52g
51g
BPF
53g
21g
Z-Δ
FIR Filter
Adaptive Update
Σ
54g
22g
50g
58g
FIR Control Filter
59g
Output Transducer
60g
57g
Σ
Training Noise
Error Path Model
42g
Σ
+
-
Error Sensor
62g
55g
64g
Update Means
61g
Error Path Model
63g
65g

Fig. 7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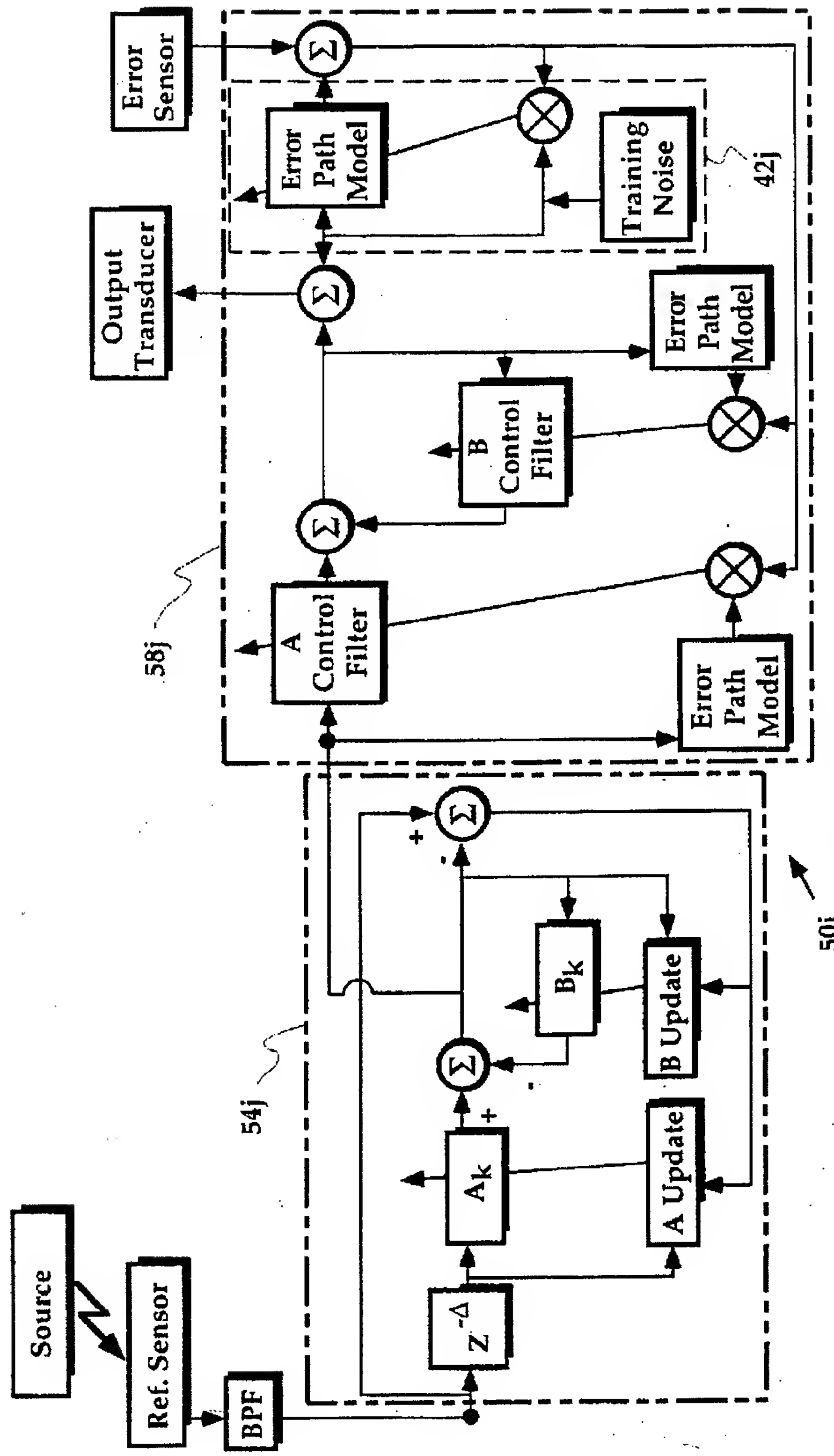
62h
Error Path Model
42h
Output Transducer
60h
Noise Source
Update Means
FIR Control Filter
58h
59h
Error Path Model
22h
54h
52h
51h
Source
Ref. Sensor
BPF
$Z^{-\Delta}$
A_k
A Update
B_k
B Update
50h

Fig. 8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Error Path Model
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A_k
A Update
B_k
B Update
54j
50j

Fig. 9

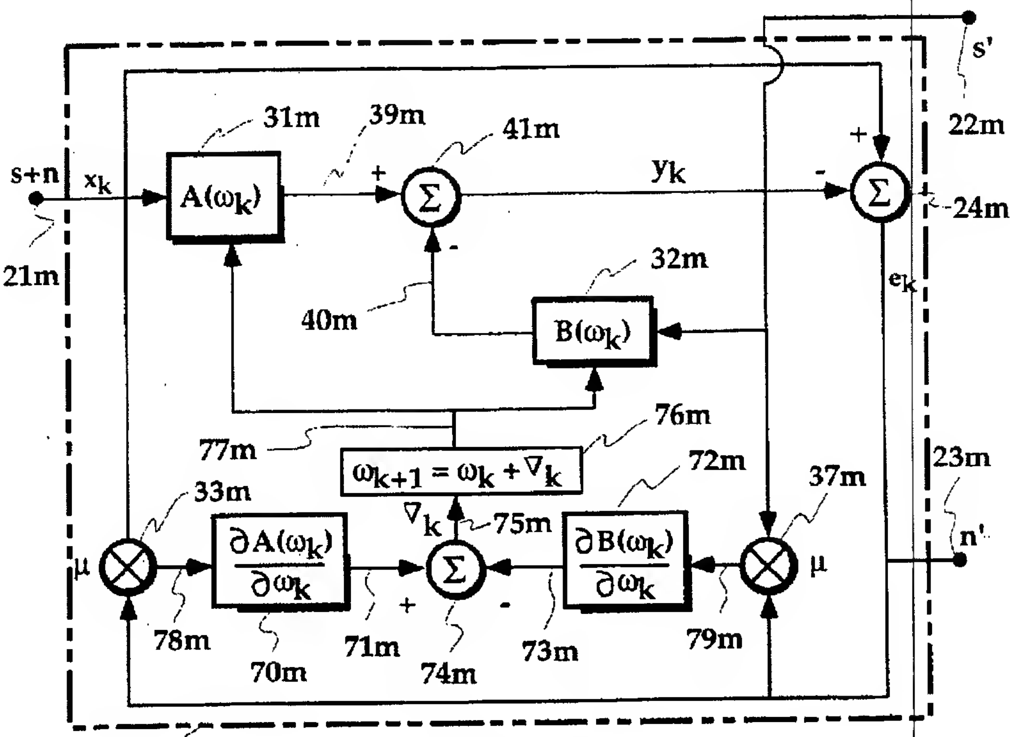
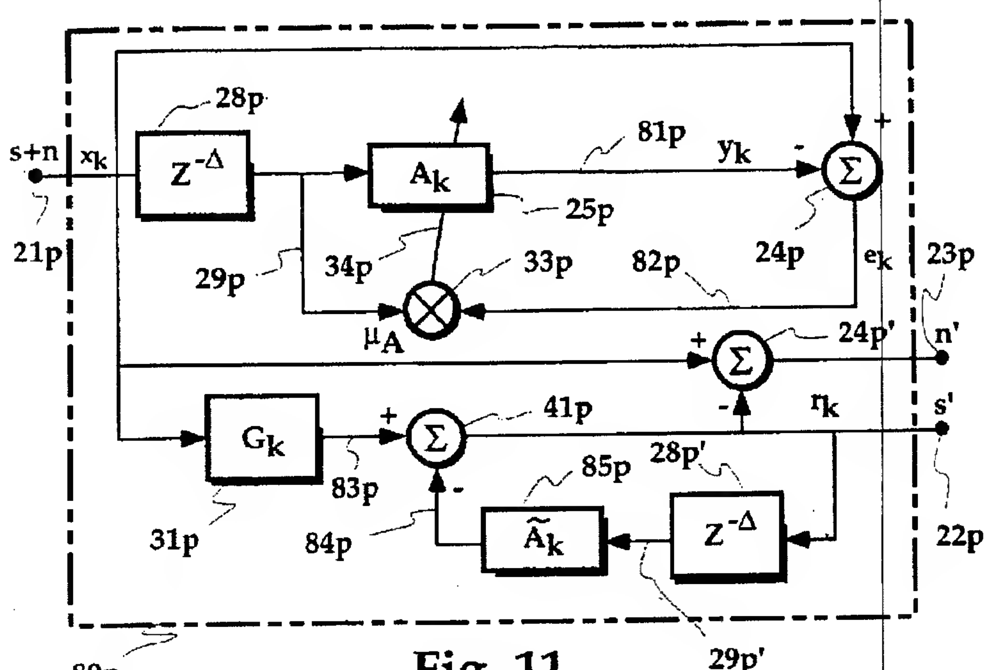

s'
22m
s+n
x_k
31m
39m
41m
$A(\omega_k)$
+
Σ
y_k
+
-
Σ
24m
21m
-
32m
e_k
40m
$B(\omega_k)$
76m
77m
$\omega_{k+1} = \omega_k + \nabla_k$
72m
23m
33m
37m
∇_k
75m
n'
μ
$\frac{\partial A(\omega_k)}{\partial \omega_k}$
Σ
$\frac{\partial B(\omega_k)}{\partial \omega_k}$
μ
+
-
78m
71m
73m
79m
70m
74m
69m

Fig. 10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81p
s+n
x_k
$Z^{-\Delta}$
A_k
y_k
-
+
Σ
25p
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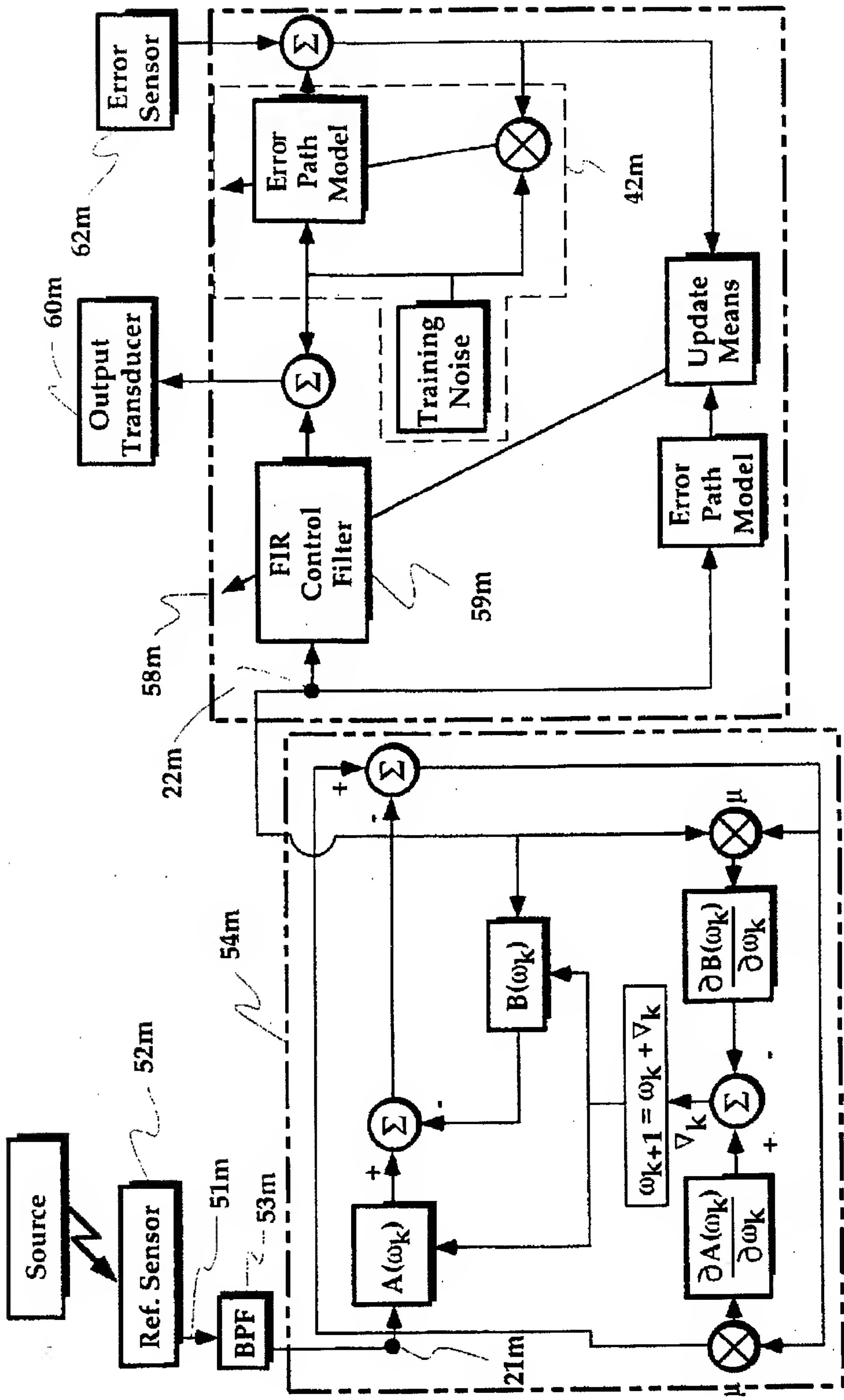
82p
24p
e_k
23p
μ_A
+
Σ
24p'
n'
G_k
+
Σ
41p
-
r_k
s'
83p
85p
28p'
-
31p
84p
$\tilde{A}_k$
$Z^{-\Delta}$
22p
80p
29p'

Fig. 11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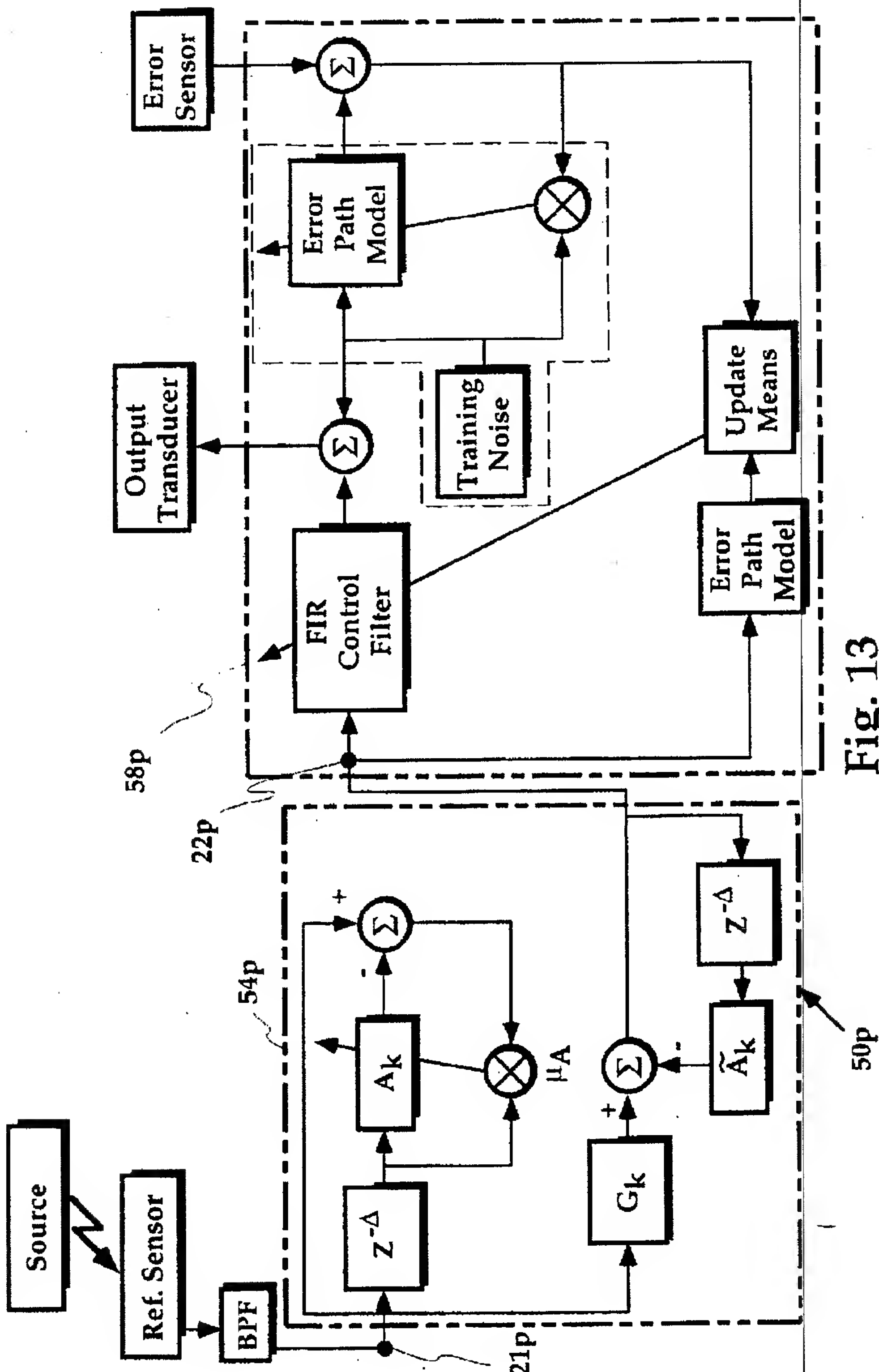
22m
Source
Ref. Sensor
BPF
52m
51m
53m
54m
21m
50m
A(ωk)
B(ωk)
Σ
Σ
Σ
ωk+1 = ωk + ∇k
∇k
∂A(ωk)/∂ωk
∂B(ωk)/∂ωk
μ
μ
+
-
+
-
+
-

Fig. 12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Training Noise
Update Means
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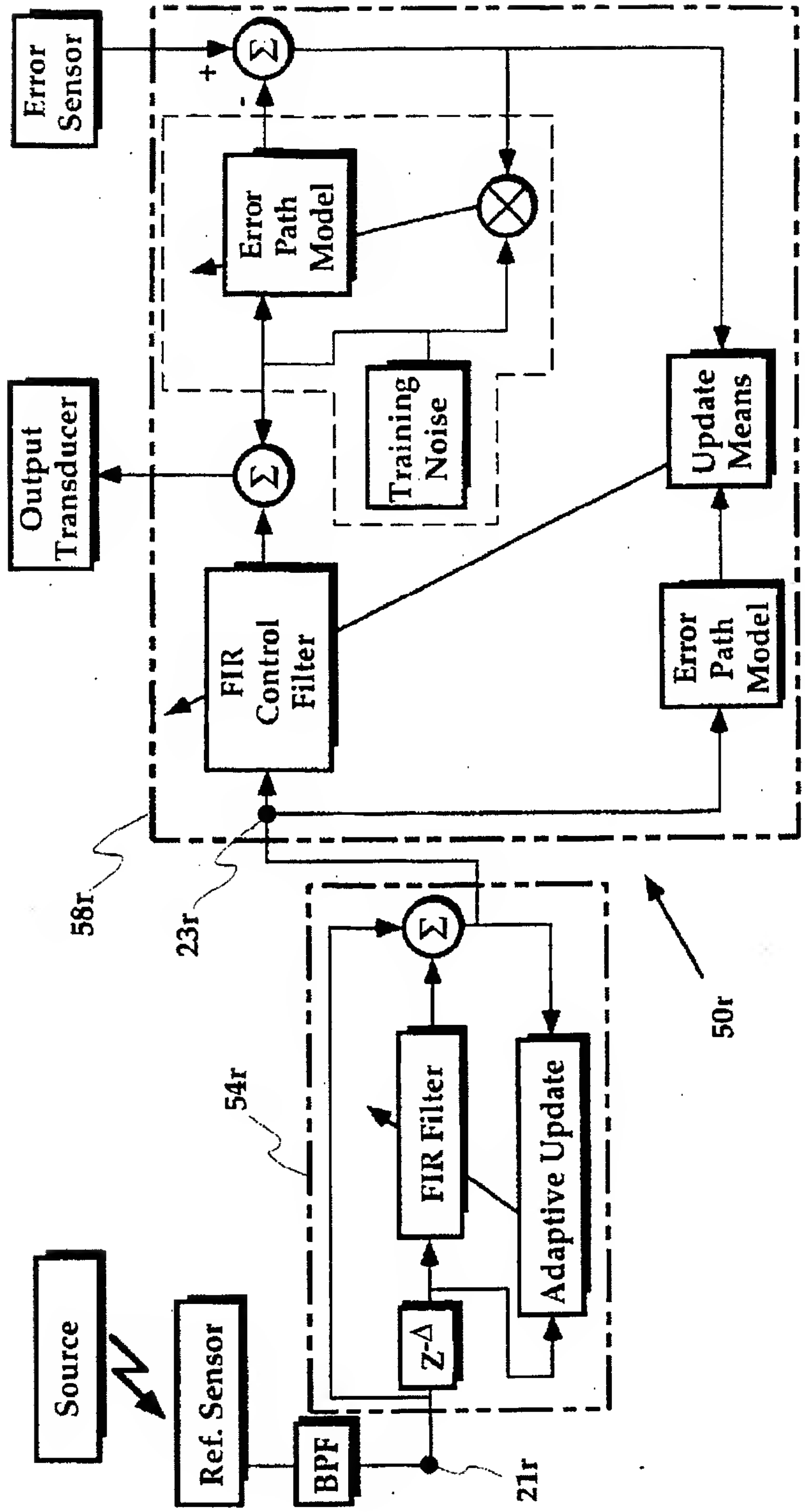
Error Path Model
58p
22p
Source
Ref. Sensor
BPF
+
-
Σ
54p
A_k
μ_A
$z^{-\Delta}$
G_k
+
Σ
-
$\tilde{A}_k$
$z^{-\Delta}$
50p
21p

Fig. 13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Error Path Model
Training Noise
Update Means
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Ref. Sensor
BPF
21r
54r
z-Δ
FIR Filter
Adaptive Update
50r

Fig. 14

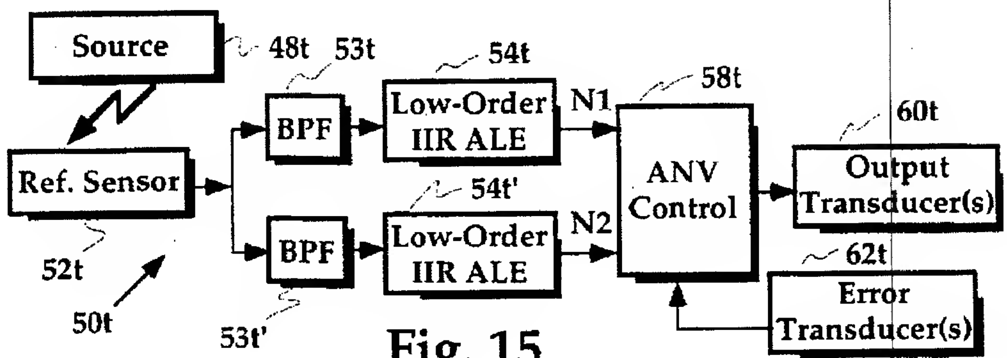


Fig. 15

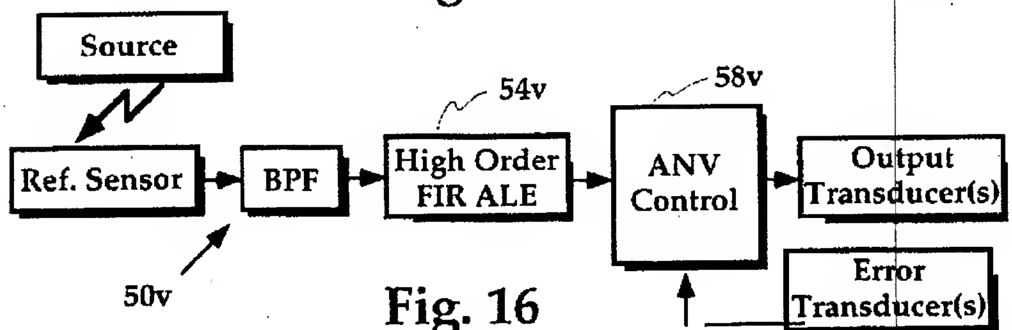


Fig. 16

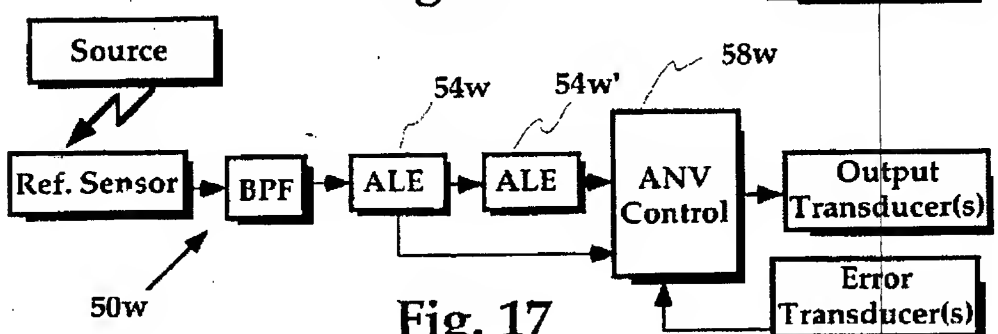


Fig. 17

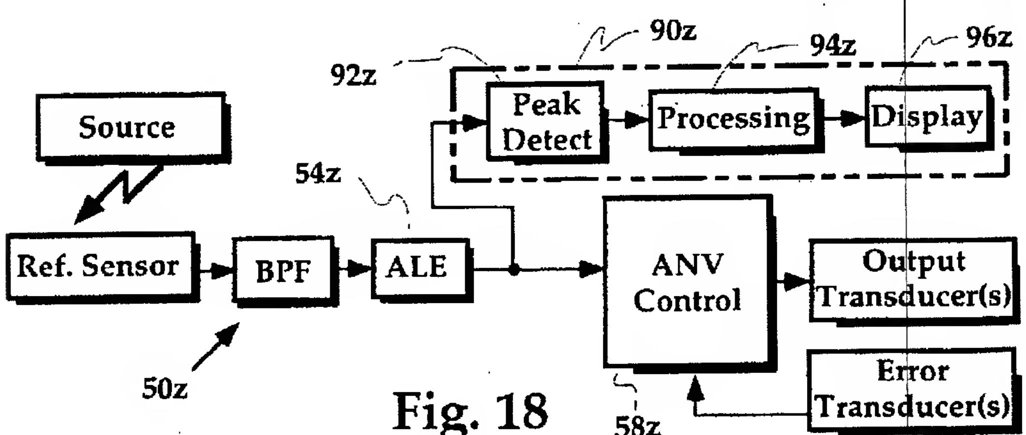
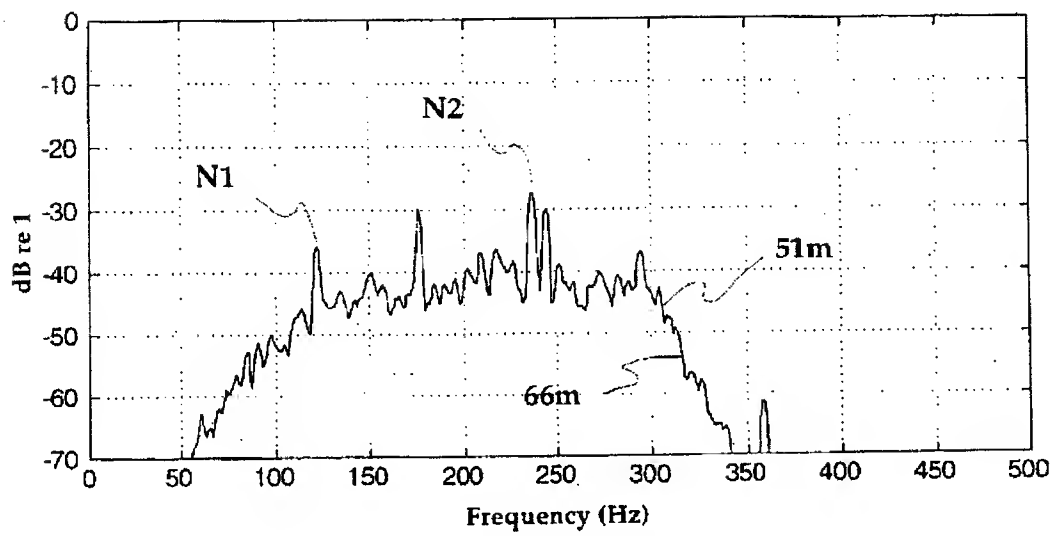


Fig. 18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-10
-20
-30
-40
-50
-60
-70
dB re 1
N1
N2
51m
66m
0
50
100
150
200
250
300
350
400
450
500
Frequency (Hz)

Fig. 19

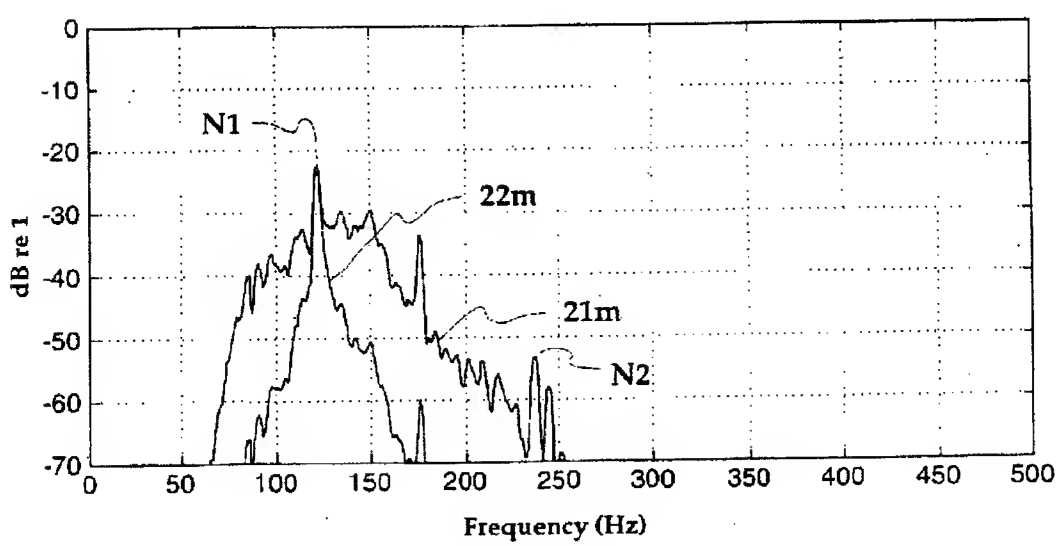
0
-10
-20
-30
-40
-50
-60
-70
dB re 1
N1
22m
21m
N2
0
50
100
150
200
250
300
350
400
450
500
Frequency (Hz)

Fig. 20

(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 0 814 456 A3

(12) EUROPEAN PATENT APPLICATION

(88) Date of publication A3:
07.10.1998 Bulletin 1998/41

(51) Int Cl.6: G10K 11/178

(43) Date of publication A2:
29.12.1997 Bulletin 1997/52

(21) Application number: 97302679.2

(22) Date of filing: 18.04.1997

(84) Designated Contracting States:
DE FR GB

(30) Priority: 17.06.1996 US 673458

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(54) **Active noise or vibration control (ANVC) system and method including enhanced reference signals**

(57) An ANVC system (50) and method of enhancing the reference (Input) signal (51) provided from a reference sensor (52) in a feedforward-type Active Noise or Vibration Control (ANVC) system (50). Preferably, an Adaptive Line Enhancer (ALE) (54) is provided in the input path for reducing the uncorrelated noise present in the reference signal (51). In one aspect, a tone(s) present can be enhanced by reducing broadband uncorrelated noise. In another aspect, the broadband input to the ANVC control (58) can be enhanced by eliminating uncorrelated tone(s). The filter structure used in the ALE (54) may include IIR or FIR and the algorithm used to update the ALE filters may include LMS, RLS, or GMV. Parametric and adaptive inverse ALEs (69) and (80) are also described. In alternate embodiments, multiple ALEs are arranged in cascaded or parallel relationship. Further, the tonal output (22) of the ALE (54) may be used as an input to auxiliary components such as Engine Vibration Monitors (EVMs). The ALEs are beneficial in both tonal and broadband ANVC systems (50).

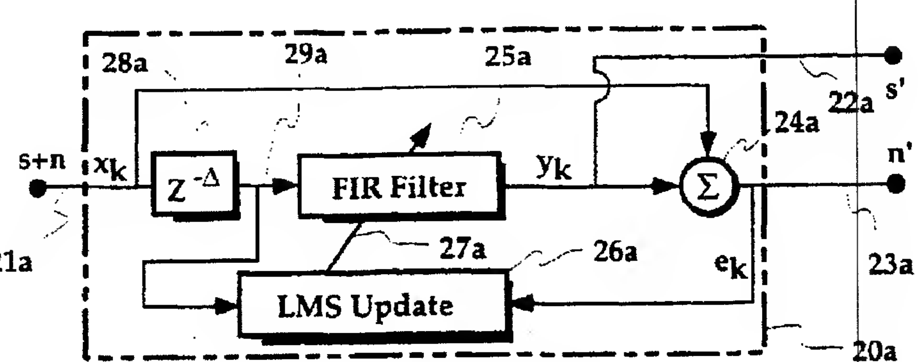

Fig. 1 Prior Art

EP 0 814 456 A3

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 97 30 2679

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.6)
X	EP 0 590 350 A (MATSUSHITA ELECTRIC IND CO LTD) 6 April 1994 * column 7, line 13 - column 8, line 14 *	1-11	G10K11/178
X	EP 0 598 120 A (FUJITSU TEN LTD) 25 May 1994 * column 7, line 5 - line 18; figure 4D *	1,2,4,6,7	
A	WIDROW B ET AL: "ADAPTIVE NOISE CANCELLING: PRINCIPLES AND APPLICATIONS" PROCEEDINGS OF THE IEEE, vol. 63, no. 12, 1 December 1975, pages 1692-1716, XP000567974 * page 1708, right-hand column, line 5 - line 31 * * page 1711, right-hand column - page 1716 *	1,7	
A	US 5 226 016 A (CHRISTMAN RUSSEL A) 6 July 1993 * abstract *	1	

TECHNICAL FIELDS SEARCHED (Int.Cl.6)

G10K

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	20 August 1998	Swartjes, H

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
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